Free Questions for 300-815 by certscare

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Question 1

Question Type: **MultipleChoice**

A company has users that are logged in to hunt groups. However, there is a requirement for hunt group configurations to provide an option to turn on audible ringtones when calls to a line group arrive at a phone that is logged out and on a break. This ringtone alerts a logged-out user that there is an incoming call to a hunt list to which the line is a member, but the call does not ring at the phone of that line group member because of the logged-out status. Which action meets this requirement?

Options:

A- Configure the HLog softkey on the phone so that while a user is logged off, it plays an audible tone when a call is missed.

B- Set the service parameter Party Entrance Tone to True.'

C- Configure the service parameter hunt group logoff notification and specify the name of the ringtone file.

D- Set the service parameter Enterprise Feature Access number for hunt group logout and set up an access number

Answer:

C

Question 2

Question Type: **MultipleChoice**

An engineer has temporarily disabled toll fraud prevention for SIP line calls on a Cisco CME12.6x and must enforce security and toll fraud prevention for the SIP line side on Cisco Unified CME. Which configuration must be used to start this process?

Options:

A- voice service volp
Ip address trusted list

B- voice service volp
enablo ip address trust authentication

C- voice service volp
enable Ip address trust list

D- voice service volp
ip address trusted authenticate

Answer:

D

Question 3

Question Type: MultipleChoice

An engineer is troubleshooting local ringback on a Cisco SIP gateway The gateway is not ignoring the SIP 180 response with SDP from the service provider, and the far end device is sending the 180 with SDP to play ringback from the IP address specified m the SDP Which configuration change must be made on the gateway to resolve the issue?

Options:

A- Router(conf-voi-serv)# dlisable-early-media 180

B- Router(conftg-sip-ua)# disable-early-media 180

C- Router(con(-voi-serv)# no disable-early-media 180

D- Router(config-sip-ua)# no disable-early-media 180

Answer:

B

Question 4

Question Type: MultipleChoice

An administrator is working on an issue between the customer s Cisco Unified Border Element and the service provider. The provider only wants to see mid-call signaling from the Cisco Unified Border Element for fax calls. Which command must be configured on Cisco Unified Border Element?

Options:

A- midcall-signallng passthru

B- midcall-signaling preserve-codec

C- no update-callerid

D- midcall-signaling passthru media-change

Answer:

D

Question 5

Question Type: **MultipleChoice**

Refer to the exhibit.

```
!
dial-peer voice 10 voip
 description Inbound
 session protocol sipv2
 incoming called-number 2000
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 20 voip
 description Outbound
 destination-pattern 2.
 session protocol sipv2
 session target ipv4:192.168.100.101
 voice-class sip options-keepalive
 dtmf-relay rtp-nte
!

CUBE#show dial-peer voice summary
dial-peer hunt 0
             AD                                  PRE PASS SESS-SER-GRP\   OUT
TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN    FER THRU SESS-TARGET     STAT PORT    KEEPALIVE    VRF
10     voip  up   up                              0  syst                                           NA
20     voip  up   up             2.               0  syst ipv4:192.168.100.101          busyout     NA
```

A call mode through the Cisco Unified Border Element to pilot 2000 is foiling. What is causing the call to foil?

Options:

A- No codecs are configured on the dial peers

B- The Cisco Unified Border Element is not receiving a response to its OPTION keepahves.

C- The destination pattern is incorrect for the dialed number.

D- VAD was not disabled on the outgoing dial poor.

Answer:

C

Question 6

Question Type: MultipleChoice

Refer to the exhibit.

```
55697959.007 |12:20:50.913 |AppInfo  |RouteListCdrc::createPartyTransformedCcSetupReqMsg
- before DAapplyCdpnXform() preXformCdpn=11112222 preTag=SUBSCRIBER prePos=11112222
crCdpnMask=33334444 crPrefixDigit= crDDI=2
55697959.008 |12:20:50.913 |AppInfo  |RouteListCdrc::createPartyTransformedCcSetupReqMsg
- after  DAapplyCdpnXform() xformCdpn=33334444 xformTag=SUBSCRIBER xformPos=11112222
55697959.009 |12:20:50.913 |AppInfo  |RouteListCdrc::transformed cdpn (without unconsumpt
digits) = 33334444, unconsumed digit=
```

Which INVITE is sent to 10.10.100.123 as a result of this log?

A)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:11112222@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "11112222" <sip:11112222@10.122.200.50>;tag=32412716~41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

B)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
message to 10.10.100.123 on port 5060 index 41
[95992364,NET]
INVITE sip:33334444@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "11112222" <sip:11112222@10.122.200.50>;tag=32412716~41f7
To: <sip:11112222@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

C)

```
55698034.001 |12:20:50.922 |AppInfo  |SIPTcp - wait_SdlSPISignal: Outgoing SIP TCP
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[95992364,NET]
INVITE sip:33334444@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
From: "1000" <sip:1000@10.122.200.50>;tag=32412716~41f7
To: <sip:33334444@10.10.100.123>
Date: Thu, 01 Apr 2021 17:20:50 GMT
Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
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INVITE sip:11112222@10.10.100.123:5060 SIP/2.0
Via: SIP/2.0/TCP 10.122.200.50:5060;branch=z9hG4bK268d6e4e48f3ae
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Call-ID: 99878a80-66100f2-265e57-67071d0a@10.122.200.50
Supported: timer,resource-priority,replaces
Min-SE:  1800
User-Agent: Cisco-CUCM12.0
```

Options:

A- Option A

B- Option B

C- Option C

D- Option D

Answer:

C

Question 7

Question Type: **MultipleChoice**

An administrator is trying to apply configuration changes on Cisco CME. When the users registered on Cisco CME to dial a local number to a PSTN call, the Cisco CME sends an incorrect number of digits. What translation rule fixes the issue and sends the correct number of digits?

Options:

A- voice translation-rule 1

rule 1 /^4...$/2404\0/ type any national plan any Isdn

B- voice translation-rule 1 rule 1 // // type any subscriber plan any isdn

C- voice translation-rule 1 rule 1 /^4...S/ /9132404 0/ type any subscriber plan any Isdn

D- voice translation-rule 1

rule 1 /^4...V /2404\0/ type any subscriber plan any isdn

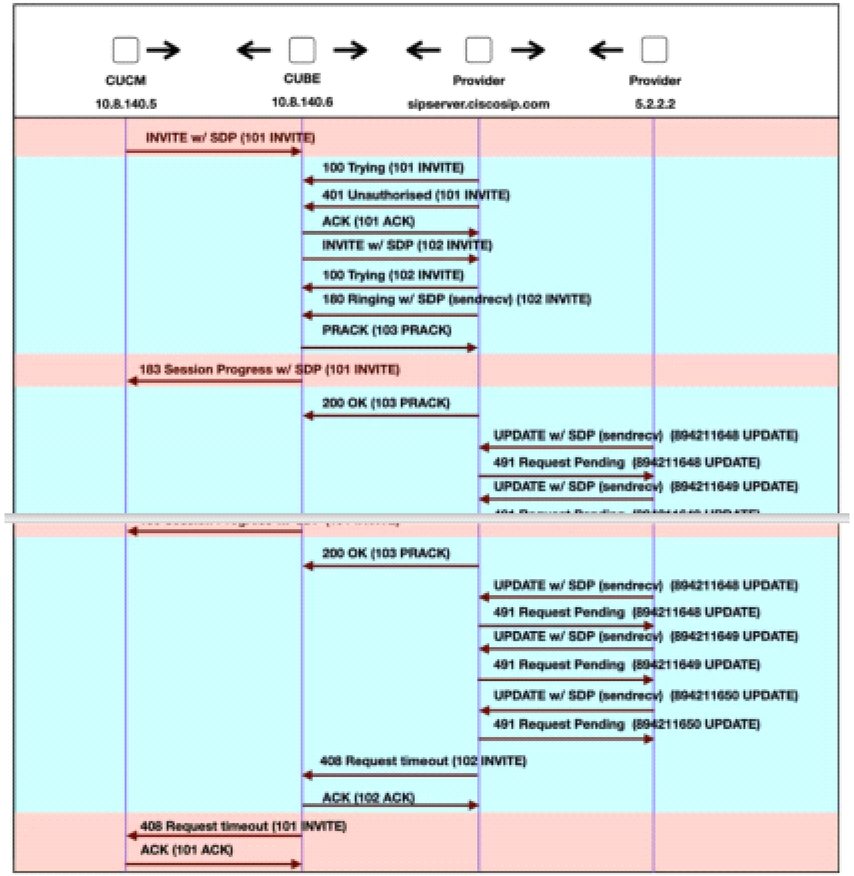
Answer:

D

Question 8

Question Type: MultipleChoice

Refer to the exhibit.



A Cisco Unified Border Element continues to send 180/183 with the required: 100rel header to Cisco UCM. and the call eventually disconnects How is the issue resolved?

Options:

A- Enable 'SIP Rel1XX Options* and -Early Offer Support' on the SIP Profile Configuration Page in Cisco UCM.

B- Enable *Early Offer support for voice and video calls' on the SIP Profile Configuration Page in Cisco UCM.

C- Disable 'SIP Rel1XX Options* and 'Early Offer Support* on the SIP Profile Configuration Page m Cisco UCM.

D- Disable 'Send send-receive SDP in mid-call INVITE* on the SIP Profile Configuration Page in Cisco UCM.

Answer:

B